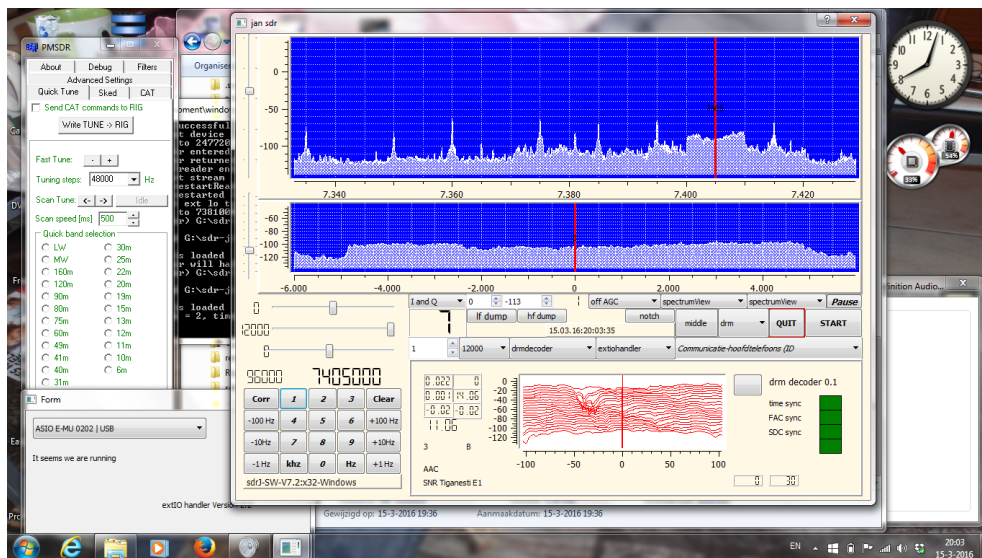


# Synchronization in the SDR-J DRM decoder\*

*an informal introduction*

Jan van Katwijk  
Lazy Chair Computing  
The Netherlands  
*J.vanKatwijk@gmail.com*

March 3, 2017



## 1 Introduction

DRM is probably not (yet?) the most popular technique for delivering audio and data over a shortwave radio channel. Nevertheless, in this time of digital modes for all kinds of data to be transferred, it is worth investigating.

SDR-J is a (hobby) project in which software for the receiver side of "SDR", i.e. Software Defined Radio, is being developed. The developed software set currently consists of three major

\*©: 2016, Jan van Katwijk, Lazy Chair Computing

<sup>1</sup>The picture shows the GUI of the SDR-J sw-receiver while receiving a transmission from Radio Romania International. OS is Windows 7, input is from a pmSDR and an external soundcard EMU 202. Note that some of the other pictures in this document were made with an older version of the sw-receiver.

programs, a DAB decoder, an FM decoder and a more general (so-called) shortwave receiver (sw-receiver).

The sw-receiver[1] started as a piece of software to control - and get some sound from - the "Elektor" SDR card [2], it developed into a system for other devices as well.

In order to be able to handle a great variety of input devices and a range of decoders, input controllers as well as decoders are implemented as *plugins*. Qt, the framework used for the development, has pretty good support for that.

A dozen decoders was implemented for the sw-receiver, apart from the - almost obvious - analog usb and lsb decoders, there are decoders for several other modes, such as rtty, psk, cw, weatherfax, etc.

Currently I am using a SDRplay device [4] and a pmSDR device [5] as standard input devices for receiving shortwave transmissions under Linux. For use under Windows, a module is available that provides support for the "extIO-XXX" [3] device handlers. In my environment, both aforementioned devices are well served using the extio-XXX device handlers.

The software is being developed under Linux and is cross-compiled for Windows using the Mingw64 toolchain. It is written in C++, and use is made of the Qt and Qwt libraries for implementing the GUI. For handling the DFT's we use of the fftw library[6] and support for making sound from the aac samples is by libfaad[7]. For feeding samples to the soundcard portaudio[8] is used, and for reading and writing ".wav" files libsndfile[9] is used.

The development of a DRM decoder turned out to be more challenging than expected beforehand, handling OFDM for DRM required quite some studying and experimenting. A reasonable complete version of the decoder is operational and included in the 7.x distribution of the sw-receiver. The version supports audio decoding, and - in a limited form - decoding of MOT data.

In this paper a very informal description of the implementation of the *synchronization* in the DRM decoder is given, since, together with the multi-layer decoding of the bits, synchronization in the receiver is the major challenge<sup>2</sup> in the development.

The objective of the paper is to provide some insight in the construction of the software to ease its comprehension. After all, the software is open source and for open source to be successful it is imperative that its sources can be understood.

In section 2 a very brief introduction to OFDM and DRM is given, in section 3 we briefly discuss the steps involved in mapping the input samples onto the DRM frames, and we give a - global - overview of the structure of the front end of the decoder.

In section 4 - the major part of this paper - we discuss the structure of the individual computations in order to synchronize and equalize.

In section 5 we conclude with some final remarks.

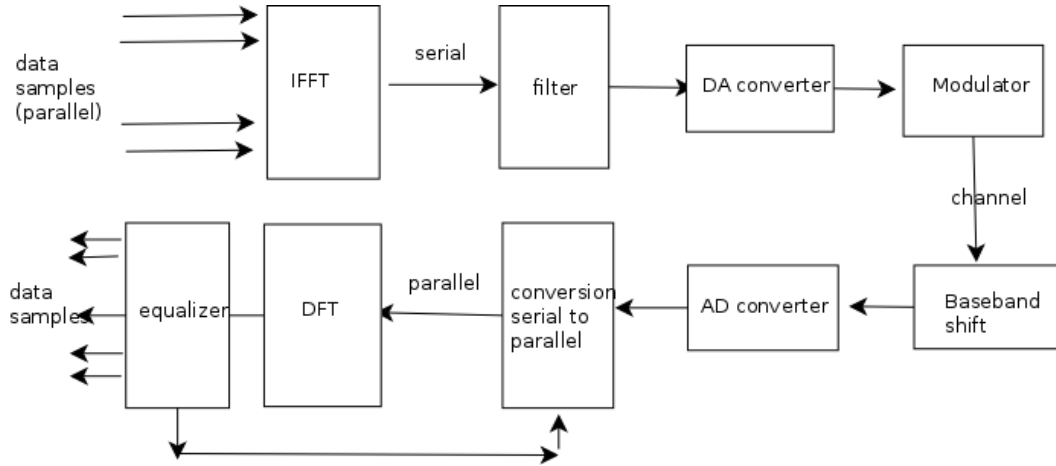
The code of the DRM decoder is to be found in the sourcetree at ".../sdr-j-swreceiver/swreceiver/plugins/decode

---

<sup>2</sup>Once the bits are "in", most work is - from a programmers point of view - straightforward and sometimes even boring

## 2 OFDM and DRM

### 2.1 OFDM



There are many introductions and tutorials on OFDM to be found on the internet e.g. [10] A (very) simplified description follows.

The principle of OFDM is pretty straightforward. Suppose we want to transmit a stream of binary words, each word containing  $N$  bits. The bits are encoded into complex values (one complex value can be used to encode up to 6 bits, using both phase and amplitude), resulting in  $M$  complex values for the encoding of the word.

This word with the  $M$  complex values is considered to represent  $M$  carrier values in the frequency domain and is fed into an IFFT device. The result is a sequence of complex values considered to be values in the time domain (symbols), i.e. the IFFT serialized the input words.

For each resulting symbol in the time domain we copy the last  $Tg$  values and add them in front of the word (a cyclic prefix) (the value of  $Tg$  is one of the key parameters). This will show to be very helpful in synchronizing the receiver with the transmitter.

The sequence of (complex) values forming a word this way then is fed into a DA converter, and the resulting analog signal is sent through some filters and a modulating device. In a continuous process such words are added to the input, through the process, to the analog output.

On the receiver side we transform the received signal into baseband, and we feed the analog signal into an AD converter to obtain a sequence of samples. We extract words of  $Tg + M$  samples, and remove in each word the  $Tg$  samples of the cyclic prefix from the word. For each word, the remaining samples are then fed into an FFT device. The result is then words with  $M$  complex values. Due to channel behaviour, the  $M$  complex values might suffer from phase and amplitude errors, however, after some beautifying the output (i.e. the equalization) we should have a pretty accurate copy of the  $M$  values that were sent. We decode the resulting  $M$  complex values from the output to get our  $N$  bits back.

The samples of the received analog data should - obviously - be fully synchronized with the samples in the sending process. Since - in general - coherent demodulation is used, the data to be decoded should be an exact copy (both phase and amplitude) of the data sent. There are many issues in reconstructing the signal, the amount of papers on topics related to this is

impressive.

## 2.2 DRM

DRM is defined in [11]

A DRM transmission is centered around transmitting *frames* as elements with relevant data.

Each frame is built up from *ofdm-symbols*, where an ofdm-symbol - we are in the frequency domain - can be considered to be built up as an array of (complex) carrier values. The amount of symbols, forming a frame is depending on the mode.

3 frames form together a *superframe*. A superframe contains audio or textual data as well as a full description of how to interpret and access the data.

Transmission - the time domain - is based on the duration of a sample of 1/12000 second. The duration of transmitting a frame is 400 msec (i.e. it takes 4800 samples), and a transmission of a whole superframe therefore takes 1200 msec.

The precise structure of a frame is defined by the *robustnessmode*. DRM supports 4 of such modes, called *A*, *B*, *C*, *D*. Different robustnessmodes have different capabilities directed towards particular transmission conditions. Robustnessmode *A* requires stable conditions with very low fading, robustnessmode *B* can handle somewhat more fading conditions etc.

Important parameters are determined by this *robustnessmode* and the *spectrum-occupancy*.

- the number of ofdm-symbols in a frame, e.g. *robustnessmode B* (used by transmissions of Radio Romania International, one of the few european transmissions using the DRM system) counts 15 ofdm-symbols in a frame, while *robustnessmode C* takes 20 ofdm-symbols with less carriers in a frame.
- The number of samples, forming a time-domain symbol is defined by variables  $T_s$ ,  $T_u$  and  $T_g$  indicating resp. the number of samples for a full time-domain symbol, the number of "useful" samples in such a symbol, and the length of the cyclic prefix (in terms of samples). For *robustnessmodes A* and *B*  $T_s$  is 320, the time it takes to transmit the data of the time-domain symbol is thus  $320 * 1/12000$  second, the time to transmit all samples constituting a frame is  $15 * 320/12000$  second, i.e. 400 msec. In *robustnessmode A* the  $T_u$  is 288, in *robustnessmode B* it is 256. The length of the cyclic prefix ( $T_g$ ) is therefore 32 samples in *robustnessmode A* and 64 in *robustnessmode B*.
- The number of carriers, depending on the *robustnessmode* and the *spectrum-occupancy*. The  $T_u$  samples in the time domain are - using an DFT - transformed into  $T_u$  carriers in the frequency domain. The outermost carriers do not contain useful data, the *spectrum-occupancy* defines which carriers of these  $T_u$  are to be used for data. As an example, the  $T_u$  of robustnessmode *B* is 256, with carriers, carrying useful data, -103 .. 103 in the *spectrum - occupancy* that is normally used.
- Pilots. Some carriers (the "pilot" carriers) contain predefined data, data on known positions in a frame that can be used for synchronization purposes. The carrier position and the contents are defined by the *robustnessmode* and the *spectrum - occupancy*. The distance (in number of carriers) between pilot carriers in an ofdm-symbol is constant.

Pilot patterns differ in subsequent ofdm-symbols, but the pilot pattern itself is repetitive, i.e. it returns every  $X$ -th ofdm-symbol within a frame. For e.g. *robustnessmode B*, the distance between successive pilot carriers in an ofdm-symbol is 6, while the repetition factor of the pattern is 3. Since there are 15 ofdm-symbols in a frame in this *robustnessmode*, each pattern appears 5 times in such a frame (note however that the pilot values on corresponding carriers in different ofdm-symbols may differ).

The basic idea is that for each carrier the channel can be constructed (and the carrier value reconstructed) by looking at the values in the pilot carriers as they appear in the received signal compared to the values as they should have been. Next to "regular" pilots, there are several other kinds of pilot:

- *time pilots* appear in the first ofdm-symbol in a frame and will be used for identifying the first ofdm-symbol in a frame.
- *frequency pilots* appear in all ofdm-symbols on the same carrier positions, positions that are defined by the *robustnessmode*. These frequency pilots may assist in determining the carrier zero. Once we know which carrier is the real carrier "zero", we can easily compute the amount of carriers that we are off.
- *boosted pilots* are pilots at carriers at either end of the ofdm-symbol with a somewhat higher amplitude than the others.

Note that the sets of pilots are not disjoint, i.e. some regular pilots are also "frequency pilots", and all boosted pilots are regular pilots.

- Data is organized in three classes:
  - a Fast Access Channel (FAC), data in a set of fixed locations within each frame that contain data, relevant for the interpretation of the data in this and other frames. The FAC cells are encoded 4QAM (i.e. 2 bits per value)<sup>3</sup> so basically as long as the phase is more or less correct, correct data can be extracted.
  - a Service Description Channel (SDC), data appearing in the first few ofdm-symbols of the first frame of a superframe. The data in the SDC is used to provide additional information on the contents of the superframe. SDC cells can be encoded in either QAM4 (with 2 bits per value) or QAM16 (with 4 bits per value), the encoding is registered in the FAC.
  - a Main Service Channel (MSC). All carriers that are not used for either FAC, SDC or pilot are available as data cell for the Main Service Channel. These data cells may contain sound or text, a description is to be found in the SDC. MSC cells are encoded as QAM16 or QAM64 (6 bits per value).

Decoding of all data items is *coherent*, i.e. for decoding both the phase and amplitude of the value are important. It is assumed that the values contained in the different carriers in the reconstructed frame resemble the originally values sufficiently to allow decoding into bits.

---

<sup>3</sup>for 4QAM the quadrant the value is in, is determining the 2 bits. For 16QAM each quadrant houses 4 possible positions, and the value can encode 4 bits determined by phase and amplitude. For 64QAM each quadrant houses 16 possible positions, and the value can encode 6 bits determined by phase and amplitude.

Obviously, 4QAM encoded data is not that sensitive to phase errors or amplitude errors. 16QAM and 64QAM require more precision in the reconstruction of the frames.

### 3 From samples to frames: global overview

#### 3.1 Introduction

Once we manage to get the frames identified and filled with the appropriately restored data values, extracting bits is pretty straightforward (multi-layer and convolutional decoding is used). The main challenge for the DRM decoder is therefore the reconstruction of the frames.

The first hurdle to overcome is to locate the position of the first sample in the time-domain inputstream and to detect the *robustnessmode* of the transmission. Contrary to e.g. DAB, the inputstream does not contain particular markers between time-domain symbols or frames, so we need to look at the data itself to find the start.

Once the location of the first sample of a time-domain symbol in the inputstream is known, the *robustnessmode* and the *spectrum – occupancy* can be computed and relevant, mode dependent, parameters, i.e. the  $Tu$ ,  $Tg$  and  $Ts$ , the number of carriers with useful data, the pilot positions within the ofdm-symbols and their pilot values, are known.

Knowing these parameters, processing a samplestream can be viewed as shifting in  $Ts$  samples per time-domain symbol (where  $Ts$  is defined by the robustnessmode of the transmission) into a buffer, deleting the  $Tg$  samples of the cyclic prefix, thus extracting the  $Tu$  samples with useful data, applying a DFT and *equalizing* the data in the ofdm-symbol such that the resulting data is - as good as possible - a copy of the data transmitted.

Building up a frame requires the identification of the first ofdm-symbol of this frame, then collecting the subsequent  $symbolsperFrame - 1$  ofdm-symbols, where  $symbolsperFrame$  denotes the amount of symbols in a frame for the mode we are in. *Time pilots* are helpful with the identification of the first symbol in a frame.

*Equalization* will - in general - require looking at the values on the pilot positions and compute the channels of the pilot positions. The equalization of the data carriers in a symbol of the frame then can be done using some form of interpolation of the transforms for the pilot carriers.

After equalization, the values of the carriers in the ofdm-symbols in the frame are assumed to be "exact" copies of the data as sent.

Each complex value of a data carrier then represents a number of bits, depending on the coding (i.e. 2 bits for QAM4, 4 bits for QAM16 and 6 bits for QAM64).

There are a few things to address though:

- as discussed, the first sample of a time-domain symbol in the input stream has to be detected;
- in general there will be some *frequency offset* in the baseband signal in the receiver side, sometimes a constant offset by some offset in the frequency selection, sometimes a varying offset due to instability of the oscillator. The frequency offset should be detected and compensated for.
- at least as important - we have to ensure that the *time offset* is compensated for. A time offset may occur since

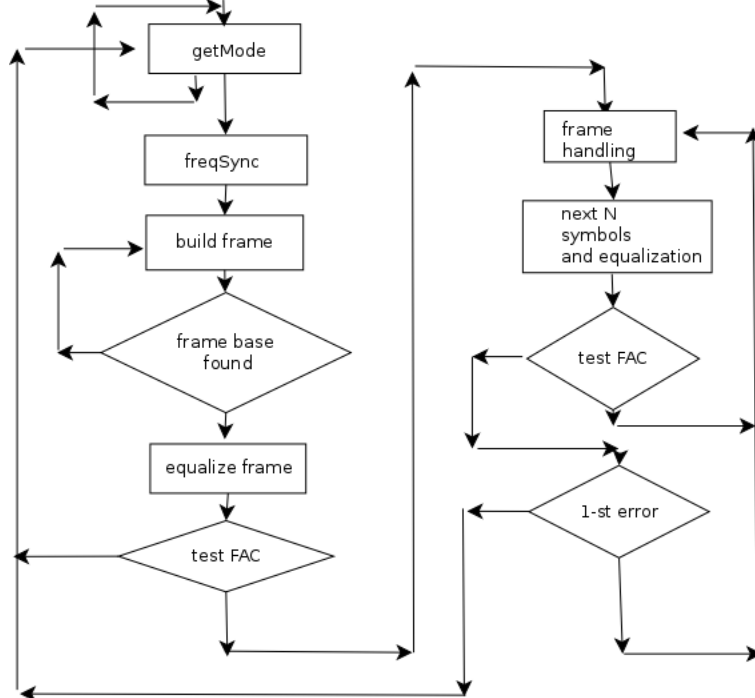
- the sampling positions in the analog signal do not necessarily coincide with the precise position the samples had in the input datastream, i.e. a timing error.
- the clock used for sampling may not be accurate.
- phaseshifts and amplitude errors due to the channel behaviour should be compensated for.

Frequency as well as time offset errors are really killing the performance, so spending some additional effort to get that right is really a must.

### 3.2 Main flow in the DRM decoder

In our DRM decoder, frame reconstruction is driven by a mainloop, continously executed in a separate thread.

In a schematical form the flow can be depicted as in the picture below.



In this picture:

- the main loop is implemented in the *run* function in the class *frameProcessor*.
- the loop starts with a call to a function *getMode*, member of the class *Syncer*. That function will *peek* into the inputstream and will (try to) compute the robustnessmode, the location of the first sample of a time-domain symbol in the inputstream, and - additionally- it will compute a (preliminary) estimate for the time offset, as well as for the (fine) frequency offset and the sampling clock offset. If not successful, the value -1 is returned. In the case of no success, the function will skip over a number of input samples and make a new attempt. If successful, the input pointer will be set to the first sample of the time-domain symbol in the inputstream.

- Once we know where the time-domain symbol starts in the inputstream, we instantiate a class *ofdmProcessor*. That class contains functions for reading samples with the given robustnessmode, and it contains the function *frequencySync*.

When *frequencySync* is called it will peek into the input (now starting at the first sample of a time-domain symbol), and it will compute the coarse frequency offset, an *integer frequency offset*, i.e. the frequency offset expressed in terms of the frequency distance between two successive carriers. The function will also try to figure out what the *Spectrum – occupancy* is by looking at the spectrum. (Unfortunately, if the *Spectrum – occupancy* found differs from the default one some instances of classes have to be deleted and reinstantiated with the correct class.)

- As soon as the *robustnessmode* and the *spectrum – occupancy* are known, all relevant parameters can be extracted. We know the *Ts*, *Tu* and *Tg* as well as the number of carriers with useful data and the carrier numbers for the lowest (*K\_min*) and the highest (*K\_max*) carrier with useful data, we know the *symbolsinFrame*, i.e. the number of symbols constituting a frame, and we start reading in *symbolsinFrame* ofdm-symbols by calling "getWord" to fill the buffer *inbank* with symbols. *getWord*, an overloaded member of "frameProcessor" will read in samples, it will do (time and frequency) adjustments, apply an FFT and select and reorder the "useful" carriers and make them available.
- We continue reading by calling this function *getWord* until we capture a *symbolsinFrame* symbols, potentially forming a frame. Since the first (zero-th) ofdm-symbol of the frame has additional pilot information, checking that we indeed have a full frame is by looking at *symbolsinFrame – 1* symbols back and correlating the samples of the symbol with the time pilots for the given robustnessmode.
- We equalize the frame by sending the *symbolsinFrame* symbols to a method *equalize*, a member of the class *Equalizer\_1*<sup>4</sup>. This equalizer needs to look into some ofdm-symbols of a next frame before having finished equalizing a frame. As soon as the equalizer finishes equalizing a frame, the equalized frame resides in a two dimensional structure *outbank*, with values for ofdm-symbol *i* at *outbank[i]*.
- ... and we test whether or not we can extract a valid FAC from the frame, a function *getFACdata*, applied to the frame, will return *true* is the FAC data meets the CRC requirements. The function is a member of the class *frameProcessor*.

If the FAC test fails, we give up and restart.

- If we do have a valid FAC, it is pretty reasonable to assume that we have a frame. We handle the frame (a description of which is not part of this paper) and interpret the data, e.g. by generating audio and extracting other data.
- After having handled the frame, we read in the next N time-domain symbols, using an overloaded (different) version of *getWord*, member of the class *ofdmProcessor*. The function will make new ofdm-symbols available. We do the equalization (*equalize*) such

---

<sup>4</sup>Many different equalizers were tested, the number "1" is reminiscent of that



that we have a next full frame. The equalizer will also return *tracking values* for the frequency offset and for the *time offset* and will recompute the sampleclock offset, these values will be used by the version of *getWord* used in this loop, to compensate the errors in time and frequency.

- Again, the basic validity test is whether or not the frame will have a valid FAC part. If the frame passes the FAC test (*getFACdata*), we continue our main loop.
- If, however, the FAC test fails, we won't be too rigid, if it is the first failure after a sequence of successes, we just consider it a mistake and continue our main loop. If it is the second error in a row, we assume a synchronization failure and give up.

### 3.3 Classes

The following classes are relevant for the synchronization

- the class *drmDecoder* is the class managing the interface to the *sw – receiver* and handling the local GUI. It will receive the incoming samples and put them into the buffer, the latter implemented in the class *simpleBuf*.
- the class *simpleBuf* implements the input buffer. The class has a member function *needed(n)* that, when called, will ensure there are "n" samples available. If needed it will wait until the amount is in the buffer (assuming the client runs in a separate thread).
- the class *Syncer*, the home of the *getMode* function inherits from the class *simpleBuf* to ease lookahead in the inputbuffer.
- the class *ofdmProcessor* implements the mappers from the samplestream in the time-domain to the ofdm-symbols in the frequencydomain. It has three overloaded members *getWord* and contains the public member *frequencySync* and the private member *get\_binZero*.
- the class *frameProcessor* implements the main loop, continuously trying to synchronize on the input and construct frames for data extraction. It is derived from the class *QThread*. It has a public member *stop* that, when called, will set a variable *isRunning* to *false*. This variable will be tested in the main loop.
- the class *Equalizer\_1* implements the equalizer. For historical reasons it is derived from a class *equalizer\_base*. *Equalizer\_1* has - apart from constructors - a single public member, called *equalize* that - when called - adds an ofdm-symbol to the equalizer. It returns *true* when a frame is synchronized, and it returns through its parameters tracking values for the fine *frequency offset*, and *time offset* and it will compute a new estimate of the sample clock offset.
- the class *Estimator\_x* with - apart from the constructors - a single public member, called *estimate* that, when presented with the values of a symbol, will return the channel estimate for the pilot carriers in that symbol. In the configuration setting of the current implementation, one may choose among three implementation for the channel estimation.

## 4 Computations for the synchronization

While many of the operations are trivial, operations dealing with synchronization, pretty crucial for successful DRM decoding, are not. Some of these operations will be discussed in more detail:

- extracting the *robustnessmode* and *spectrum – occupancy* and locating the start of the symbol in the time domain stream.
- initial computation of the "coarse" frequency offset for correcting the VFO, and
- initial computation of the "fine" frequency offset for correcting the VFO and the time offset.
- Once we are in sync and running, the offsets of the fine frequency offset and time offset are continuously computed and used for compensation.
- channel estimation and equalization. After applying the DFT we equalize the ofdm-symbols, such that we have the right (or more realistic: best) values for bit decoding.

### 4.1 Extracting the robustnessmode and locating symbol start

The robustnessmode can only be detected by looking at the samples in the time domain inputstream. The basic idea is to look at correlations: when we know the location of the first sample of a time-domain symbol, we also know that the sum of the correlations of the first  $Tg$  samples with the corresponding samples  $Tu$  places ahead is maximal. This follows directly from the use of the cyclic prefix, which is (at least at transmission time) an exact copy of the samples at locations  $Tu..Ts$  within the time-domain symbol.

A first approach then would be to compute the correlation for a sequence of samples for a given robustnessmode. If  $z$  is the inputstream we compute for a sample with index  $m$ :

$$\Phi(m) = \text{abs}(\sum_{r=0}^{Tg-1} z_{m+r} \times z_{m+Tu+r}^*)$$

and look for the index  $m$  where  $\Phi(m)$  is maximal.

In the DRM decoder we apply an MSEE based extension (from [12], formula 5.16)<sup>5</sup> to compute  $\Phi$  for a given index  $m$  in the input stream  $z$ .

$$\Phi_{msse}(m) = \sum_{r=0}^{Tg-1} \text{abs}(z_{m+r}^2) + \sum_{r=0}^{Tg-1} \text{abs}(z_{m+Tu+r}^2) - 2 \times \text{abs}(\sum_{r=0}^{Tg-1} z_{m+r} z_{m+Tu+r}^*)$$

We then compute  $\Phi_{msse}(m)$  for  $m = k..k + Ts$  and look for the  $m$  where  $\Phi_{msse}(m)$  is minimal ( $k$  being the index of the first available sample in the inputstream).

Doing so for each of the DRM robustnessmodes considered, the minimum of the minima indicates the robustnessmode and the index of the first time-domain sample in the input stream.

**Implementation** The functionality is implemented in the function *getMode*, a member of the class *Syncer*. The class extends the class *simpleBuf* that implements the input buffer, so from *getMode* we have direct access to all buffer elements.

*getMode* computes, apart from the robustnessmode, the index in the input buffer for a symbol to start, and it computes an estimate for the fine time offset, the frequency offset and the sample clock offset.

---

<sup>5</sup>complex  $z_m^2$  is computed as  $z_m \times z_m^*$

Implementation is in two parts, in the first part the robustnessmode is determined, in the second part, relevant parameters for the mode are computed.

Implementation of the first part is an almost literal translation of the formula discussed above.

For each of the DRM robustnessmodes correlations  $s[i] \times s[i + Tu]^*$  are computed for the samples  $s[i], i \in 0..nSamples - Tu$ .

Then *summedCorrelations* and *summedSquares* are computed over the sequence of samples as sum of *Tg* correlations resp. squares.

$$summedCorrelations = \sum_{i=0}^{nSamples-Tu} s[i] \times s[i + Tu]$$

and

$$summedSquares = \sum_{i=0}^{nSamples-Tu} real(s[i] \times s[i]^* + s[i + Tu] \times s[i + Tu]^*)$$

Furthermore, we compute *Ts* terms

$$squareTerm[i] = \sum_{j=0}^{nSymbols} 0.5 \times summedSquares[j * Ts + Tg + i]$$

We store the value for the *summedCorrelations* as  $\gamma$ . The minimum value for the result of  $squareTerm[i] - 2 * abs(\gamma[i])$  is determined. For each of the 4 robustnessmodes this is done, the minimum value indicates the mode we were looking for.

A simple "validation" test is looking at the relative value of the correlation. We assume we are correct when the value obtained for the the mode found is at least double of the values for the other modes.

Computing the terms over more than a single symbol in the inputstream allows us to give rough estimates of the time offset and the sample clock offset.

For estimating the *time offset* we take the offsets (*offset<sub>i</sub>*) of the *s* subsequent symbols that we computed anyway, and apply a form of averaging[12]:

$$\frac{((\sum_{i=0}^s offset_i) \times (\sum_{i=0}^s i \times i) - \sum_{i=0}^s i * \sum_{i=0}^s i \times offset_i)}{((s-1) \times \sum_{i=0}^s i \times i - \sum_{i=0}^s i \times \sum_{i=0}^s i)}$$

Computation of the sample clock offset is done by looking at the slope of the offsets found[12]

$$\frac{((s-1) \times \sum_{i=0}^s (i \times offset_i) - \sum_{i=0}^s i * \sum_{i=0}^s offset_i)}{((s-1) \times \sum_{i=0}^s (i \times i) - \sum_{i=0}^s i \times \sum_{i=0}^s i)} \times \frac{1}{Ts}$$

Finally, computation of a first estimate of the fine *frequency offset* is simply by looking at the phase of the sum of the correlations

$$arg(\gamma[offset_{sample_0}])$$

from which it is easy to derive an estimate for the offset in Hz.

**Results** One shouldn't overrate the quality of the estimation procedure, estimates are just estimates. In the tables below we give the results of estimating the robustnessmode and the start of the first symbol in the inputstream with resp. 5, 15 and 25 time-domain symbols lookahead (note that 25 time-domain symbols in robustnessmode *A* or robustnessmode *B* means 8000 samples, which takes 2/3 of a second).

The experiment is carried out by running the DRM decoder on a prerecorded transmission with medium quality reception (i.e. pretty large parts of the transmission cannot be converted into audio). After starting the transmission and manually selecting the subband, the decoder is started.

The first estimate (i.e. "found 1") is obtained looking at the input at an unknown position. For the subsequent estimates, the buffer pointer is shifted  $Ts/2$  samples. I.e., iff the first estimate is accurate, then the next estimates should all be exactly  $Ts/2$ , for the robustnessmode under consideration this means 160 samples.

```
found 1: Mode = 2, time_offset = 288.166687, sampleoff = 0.001563 freqoff = -1.787721
found 2: Mode = 2, time_offset = 158.666687, sampleoff = 0.003125 freqoff = -1.768491
found 3: Mode = 2, time_offset = 162.833313, sampleoff = -0.001563 freqoff = -1.741446
found 4: Mode = 2, time_offset = 157.500000, sampleoff = 0.004688 freqoff = -1.754578
found 5: Mode = 2, time_offset = 162.833313, sampleoff = -0.007812 freqoff = -1.735270
```

Looking at a lookahead of 15 time-domain symbols, we get

```
found 1: Mode = 2, time_offset = 161.208801, sampleoff = -0.000309 freqoff = 0.281117
found 2: Mode = 2, time_offset = 158.450562, sampleoff = 0.000567 freqoff = 0.285227
found 3: Mode = 2, time_offset = 160.384613, sampleoff = 0.000841 freqoff = 0.285681
found 4: Mode = 2, time_offset = 158.791199, sampleoff = 0.001631 freqoff = 0.266741
found 5: Mode = 2, time_offset = 160.615387, sampleoff = 0.001202 freqoff = 0.277272
```

Looking at a lookahead of 25 time-domain symbols, we get

```
found 1: Mode = 2, time_offset = 26.641296, sampleoff = -0.000577 freqoff = -1.728946
found 2: Mode = 2, time_offset = 159.184784, sampleoff = -0.000374 freqoff = -1.728744
found 3: Mode = 2, time_offset = 160.597839, sampleoff = -0.000442 freqoff = -1.726560
found 4: Mode = 2, time_offset = 159.358704, sampleoff = -0.000374 freqoff = -1.728518
found 5: Mode = 2, time_offset = 160.163025, sampleoff = -0.000244 freqoff = -1.728915
```

For a transmission with an excellent reception, we get using a lookahead of 5 time-domain symbols

```
found 1: Mode = 2, time_offset = 103.166687, sampleoff = 0.001563 freqoff = 0.547337
found 2: Mode = 2, time_offset = 161.000000, sampleoff = 0.000000 freqoff = 0.553580
found 3: Mode = 2, time_offset = 160.000000, sampleoff = 0.000000 freqoff = 0.556413
found 4: Mode = 2, time_offset = 160.000000, sampleoff = 0.000000 freqoff = 0.558379
found 5: Mode = 2, time_offset = 160.000000, sampleoff = 0.000000 freqoff = 0.552505
```

Using lookead of 15 time-domain symbols we get

```
found 1: Mode = 2, time_offset = 183.824158, sampleoff = -0.000069 freqoff = -1.465193
found 2: Mode = 2, time_offset = 159.329651, sampleoff = 0.000309 freqoff = -1.466084
found 3: Mode = 2, time_offset = 160.175842, sampleoff = 0.000429 freqoff = -1.467475
found 4: Mode = 2, time_offset = 160.175842, sampleoff = 0.000429 freqoff = -1.469134
found 5: Mode = 2, time_offset = 160.175842, sampleoff = 0.000429 freqoff = -1.468285
```

and using a lookahead of 25 time-domain symbols we see

```
found 1: Mode = 2, time_offset = 223.554321, sampleoff = 0.000127 freqoff = 2.689186
found 2: Mode = 2, time_offset = 159.478271, sampleoff = 0.000161 freqoff = 2.689304
found 3: Mode = 2, time_offset = 160.489136, sampleoff = 0.000157 freqoff = 2.689111
found 4: Mode = 2, time_offset = 160.597839, sampleoff = 0.000151 freqoff = 2.690619
found 5: Mode = 2, time_offset = 159.804352, sampleoff = 0.000105 freqoff = 2.690310
```

The numbers do give some confidence in the approach (Note that any value  $X$  in the range  $N - 0.5 .. N + 0.5$  is considered to be an integral value  $N$  together with a fractional part ( $X - N$ )), most of the values above have an integral part of 160.

An error in the integral estimate of the position will lead to errors in the FAC decoding and therefore to restarting the process.

## 4.2 frequencySync: Computing the coarse frequency offset

For computing the coarse frequency offset we will use the fact that symbols in the frequency pilots carry three pilots with - determined by the *robustnessmode* - predefined locations and values.

Given that we know the *robustnessmode*, and the location of the first sample of the symbol in the samplestream, it is easy to read in  $Ts$  samples, and take the FFT of the last  $Tu$  samples. By taking a sequence of  $N$  of such time-domain symbols, adding up the relevant values of the FFT, we might eliminate some noise and fading effects (recall that frequency pilots occur in all symbols on the same positions).

Then, if  $s$  represents the presumed positions of carrier *zero* in the FFT output, the maximum of

$$F(s) = \sum_{n=0}^{Upperbound} \sum_{j=1}^3 abs(carrier_{n,s+pilot_j})$$

will give us the carrier  $s$  which is most likely the carrier zero. *Upperbound* indicates the number of ofdm-symbols used in the summation, in the implementation we use the number of ofdm-symbols in frames of this mode as upperbound.

Given that the (complex) samples in our implementation are taken at 12K, and given that the bandwidth of the signal can be 10K, it is obvious that the frequency range we look into is limited. Frequency errors of app more than 700 Hz are beyond (automatic) correction. Since, however, the user has a large screen, showing a.o. the spectrum, he is able to do some coarse correction by himself<sup>6</sup>.

**Implementation** The function *frequencySync* is implemented as member of the class *ofdmProcessor*. It calls upon the function *getoffsetsint* to actually compute the frequency offset and it will estimate the Spectrum-occupancy.

At first it will ensure that sufficient data for  $N$  ofdm-symbols (+ some more) are available in the input buffer. Next, it will call upon a private version of *getWord* to actually read in  $N\_symbols$  time-domain symbols and transform these into  $N\_symbols$  ofdm-symbols.

The version of *getWord* is private a.o. since it only peeks into the input stream and will not change the buffer pointer for the inputstream.

The function *get\_zeroBin* will - using the data read in into the *symbolBuffer* - decide where the carrier "zero" is, i.e. determine the frequency offset. The result is a *binNumber* in the range  $-Tu / 2 .. Tu / 2$ .

Spectrum-occupancy is estimated by peeking into a number of carrier positions (the implementation is a rewrite of the one in the software from [15]). For the computation, the *binNumber* is translated into a number in the range  $0 .. Tu$ .

For each of the possible spectra we identify the minimum and the maximum of the carriers with useful data (which follows from the *K\_min* and *K\_max* for the combination of the *robustnessmode* and the *spectrum - occupancy* we are looking at) for each of the spectra possible, and we look at carrier values nearby the edges inside and outside the relevant parts of the spectrum under consideration.

We compute the (relative) energy by comparing the energy within a *bin* that is supposed to be just inside the relevant part of the spectrum with the energy within a bin supposed to

---

<sup>6</sup>Although too large an offset cannot be compensated for automatically, the offset is shown on the screen and can be corrected manually

be (just) outside that part of the spectrum.

The maximum of the measured values indicates the Spectrum-occupancy.

*get\_zeroBin* The function *get\_zeroBin* will decide which *bin* in the collected ofdm-symbols that are stored in *symbolBuffer* represents the carrier zero.

A viable alternative to looking at the energy in the bins is to make use of the strong correlation that exists between frequency pilots in successive ofdm-symbols vs the correlation between random values.

Computing  $K(n, s)$  for symbol  $n$  ( $n > 0$ ) in the *symbolBuffer*, and for carrier  $s$  that are candidate carrier zero.

$$K(n, s) = \text{abs}(\sum_{j=1}^3 \text{carrier}_{n,s+\text{pilot}_j} \times \text{carrier}_{n-1,s+\text{pilot}_j}^*)$$

where  $\text{pilot}_j$  indicates the index for the carrier of  $j$ -th frequency pilot. Extended over the whole *symbolBuffer* this reads

$$F(s) = \sum_{n=1}^{n_{\text{Symbols}}} \text{abs}(\sum_{j=1}^3 \text{carrier}_{n,s+\text{pilot}_j} \times \text{carrier}_{n-1,s+\text{pilot}_j}^*)$$

The value  $F(s)$  for carriers  $s$  in the range  $-Tu/10..Tu/10$  is computed, the  $s$  for which the value is maximal gives the bin number. This bin number is returned.

#### 4.2.1 Results

The computations - with energy levels in the bins in computing the maximum, or with correlations in computing the maximum - depend on the quality of the signal, there is no guarantee that in all cases the right bin is designated the bin representing carrier zero.

### 4.3 Initial estimation of time and frequency offset

As long as we are not synchronized, we have to use time domain data to estimate time and frequency offset. It is assumed that the integer frequency offset is stable, i.e. we do not need to recompute it (obviously until needed).

As long as we are *not* synchronized, an overloaded version of *getWord* is used to read in time domain samples and extract ofdm-symbols, applying time domain and frequency domain corrections on the input.

#### 4.3.1 Time offset

As mentioned above, determining the Robstnessmode has - as a by-product - an estimate of the time offset. Since it is pretty hopeless to estimate time offset using only a single ofdm-symbol, we use the rough estimate from the robustness detection. For each symbol read, we adjust the time offset

$$\text{timeoffset}_{\text{symbol}+} = Ts \times \text{timeoffset}_{\text{sample}}$$

*Compensating* the time offset is done simply by linear interpolation (the time offset is smaller than half the time between two successive samples).

### 4.3.2 Fine frequency offset

In case the frequency offset is 0.0, there will be no phase difference between the samples in the cyclic prefix and the corresponding samples in the data segment, i.e.

$$\forall i \in 0..Tg.arg(s[i] \times s[Tu + i]^*) = 0$$

In case of a non-zero fine frequency offset, the correlation between the time-domain sample and the conjunct of its corresponding element in the data segment

$$arg(s[i] \times s[Tu + i]^*) \neq 0.$$

The corner itself is a reasonable indicator for the VFO offset.

By using more than a single pair of samples and averaging the value, we eliminate some undesired effects coming from noise. I.e. the fine frequency offset is computed from

$$vfoffset = \frac{arg(\sum_{i=0}^{Tg-1} s[i] \times s[Tu+i]^*)}{(2 \times \pi)} \times \frac{Samplerate}{Tu}$$

Compensating the frequency error in the inputstream is done in the time domain: a table driven "local oscillator" with a granularity of 0.01 Hz (i.e. a table with  $12000 \times 100$  elements) is used, and all samples are mixed with values from this oscillator.

$$theAngle = 0.9 \times theAngle + 0.1 \times vfoffset$$

$$theOffset = \frac{theAngle * 1200000}{2 * \pi * Tu}$$

the result is used in the local oscillator to adjust the VFO frequency in the time domain.

**Implementation** The implementation of the computation of the fine frequency offset and the correction on the time-domain samples is in the function *getWord*, member of the class *ofdmProcessor*.

## 4.4 Identifying the start of a frame

To assist in locating a frame in DRM, the zero-th ofdm-symbol of a frame contains a pretty large number of additional pilots, the *time* pilots. Identification of the beginning, i.e. locating the zero-th time-domain symbol of the frame takes place in the *run* function - implementing the thread - in the class *frameProcessor*. It is simply done by locating the symbol with the maximum correlation in a frame.

We compute

$$\forall k \in 0..symbolsinFrame \ S(k) = \sum_{i=first}^{last} carrier_{k,pilot_i} \times table_{Mode,i}^*$$

for the last N ofdm-symbols read, and return *true* if  $S_0$  is maximal. Note that all symbols of the frame are read already, so the full frame is available. (It is assumed that  $carrier_{k,pilot_i}$  indicates the value of the carrier of the  $i$ -th pilot in the  $k$ -th symbol, and *table* is an array containing the time pilot values for the various robustnessmodes in an ordered way)

Once the ofdm-symbol "zero" is detected, we feed the frame - ofdm-symbol by ofdm-symbol - to the equalizer and, after having an "equalized" frame, we verify that the data in the FAC is valid.

**Implementation** The correlation

$$S(k) = \sum_{i=first}^{last} carrier_{k,pilot_i} \times table_{Mode,i}^*$$

is implemented in a function *getCorr*, private to the class *ofdmProcessor*.

As long as we are *not* synchronized, the correlation is computed and stored in a row of *symbolsinFrame* elements.

## 4.5 Tracking the fine frequency and time offset

Once we have reasonable values for the parameters and we are in sync, i.e. the FAC values make sense, we will use the information extracted from the ofdm-symbols to compute tracking values for the time and frequency offset. Leading principle here is

- a *frequency offset* leads to phase offsets in carriers with the same index in subsequent symbols, while
- a *time offset* leads to phase offsets in carriers with increasing indices within the same symbol.

As a consequence, the computation of a tracking value for the frequency offset requires looking at more than a single symbol, while for the computation of the tracking value for the time offset, we can just look at the most recent one.

Computing the tracking values is done in the function *equalize*, member of the class *Equalizer\_1*. The input to that function is a symbol in the frequency domain, its output indicates whether or not a complete frame was equalized or not. Through its parameters the function returns three values:

- an estimate of the error in the fractional time offset,
- an estimate of the fine (i.e. fractional) frequency offset, and
- an estimate of the sample clock offset.

These values, when computed, are handed over to an (overloaded version of the) function *getWord* in the class *ofdmProcessor*.

As long as we are synchronized, the loop in the *run* function in the class *frameProcessor* contains a sequence in which *getWord* and *equalize* are called one after the other.

### 4.5.1 Tracking the fine frequency offset

All ofdm-symbols in a DRM frame carry three frequency pilot carriers, it is therefore tempting to use the phase difference between corresponding frequency pilots in subsequent symbols as basis for estimating the fine frequency offset.

On the other hand, there is a repetitive pilot pattern within the ofdm-symbols in a frame, i.e. each pilot pattern reappears after a number (defined by the parameters of the robustnessmode) of ofdm-symbols. Since there are far more regular pilots in an ofdm-symbol than frequency pilots, we can also base the computation of the tracking value of the fine frequency offset on a comparison of the data using full pilot patterns. We look at

$$\delta_f = \sum_{j=0}^{J-1} \frac{\theta_j}{(2 \times \pi \times \frac{T_s}{T_u} \times J \times d)}$$

where the  $\theta_j$  indicates the phase difference - due to a VFO offset - of pilots at the same positions in two ofdm-symbols with distance  $d$ .  $J$  is the number of pilots in the symbol.  $\theta_j$  is the phase



difference:

$$\theta_j = \arg(Z_{k,j} \times Z_{k-d,j}^*)$$

where

$$Z_{k,j} = \text{carrier}_{k,j} \times \text{pilotValue}_{k,j}^*$$

i.e. the correlation between the value that we got with the value that it should be (Note that while the pattern of the pilot will repeat itself, the pilot values for pilots on corresponding positions in different symbols will be different).

Alternatively, as stated earlier, we could be using the phase difference between corresponding frequency pilots in subsequent ofdm-symbols to compute an average offset (expressed as angle).

$$\delta_f = \frac{\sum_{k=0}^3 \frac{\arg(\sum_{j=1}^{N-1} \theta_{j,k})}{N-1}}{3}$$

where

$$\theta_{j,k} = \text{carrier}_{j,\text{freqPilot}_k} \times \text{carrier}_{j-1,\text{freqPilot}_k}^*$$

**Implementation** Estimation of the offsets is done in the function *equalize*, member of the class *Equalizer\_1*. The implementation of the two alternatives results in setting a value for "offs1" and "offs7". One of these, depending on the setting of the comments, is given back as output.

*Correction* of the detected offset is done in the function *getWord*, an overloaded member of the class *ofdmProcessor*. Correction is done by adding (a fraction of) the error value to the frequency corrector, i.e.

$$\text{theAngle} = \text{theAngle} + 0.1 \times \text{delta}_f,$$

and computing the term with which the VFO (in steps of 0.01Hz) needs to be corrected

$$\text{offset} = \frac{\text{theAngle} \times 1200000}{(2 \times \pi \times Tu)}$$

**Results** We ran some tests to look at the differences in computed angles when using different methods. The first table shows the differences in the computed offset (in Hz) when using correlation of the time domain samples (column "initial") compared to computing a tracking value using the correlation between corresponding pilots in different ofdm-symbols.

```
found 1: Mode = 2, time_offset = 155.186829, sampleoff = 0.000343 freqoff = -1.035243
.....
initial -> -7.610438, tracking -> -6.872640, error -0.069466
initial -> -7.647318, tracking -> -6.869167, error 0.064016
initial -> -7.655280, tracking -> -6.872368, error -0.193599
initial -> -7.656196, tracking -> -6.862688, error 0.036547
initial -> -7.622445, tracking -> -6.864515, error 0.071519
initial -> -7.605674, tracking -> -6.868091, error 0.027857
initial -> -7.661390, tracking -> -6.869484, error 0.102430
initial -> -7.726436, tracking -> -6.874605, error 0.028973
initial -> -7.810171, tracking -> -6.876053, error -0.003366
initial -> -7.804359, tracking -> -6.875885, error 0.080864
initial -> -7.833959, tracking -> -6.879929, error -0.056720
initial -> -7.848853, tracking -> -6.877092, error 0.080997
initial -> -7.853055, tracking -> -6.881142, error 0.061169
```

```

initial -> -7.737529, tracking -> -6.884200, error 0.001444
initial -> -7.687623, tracking -> -6.884273, error 0.067440
initial -> -7.696682, tracking -> -6.887645, error 0.019243

```

It shows that the average difference is app 0.8 Hz, for this transmission the distance (in Hz) between carriers is 46.875, 0.8 Hz is less than 2%.

The second table shows the - relative - differences between frequency offsets computed by different approaches. The "freq pilots" column contains the tracking values using the frequency pilots of the last *symbolsinFrame* ofdm-symbols, the column "all pilots" contains the values when using the phase differences between corresponding pilot carriers in successive symbols with the same pilot layout.

```

found 1: Mode = 3, time_offset = 81.345032, sampleoff = 0.000021 freqoff = 2.722603
freq error: freq pilots = -0.001238, all pilots = -0.022117
freq error: freq pilots = -0.002598, all pilots = -0.029494
freq error: freq pilots = -0.007710, all pilots = -0.068261
freq error: freq pilots = -0.004116, all pilots = -0.067595
freq error: freq pilots = -0.010344, all pilots = -0.071337
freq error: freq pilots = 0.009163, all pilots = -0.033330
freq error: freq pilots = -0.003729, all pilots = -0.054826
freq error: freq pilots = -0.006766, all pilots = -0.071762

```

However, the tracking mechanism ensures that integrating the error - computed through looking at the phase differences of frequency pilots in subsequent symbols - into a frequency offset shows reasonable results.

```

found 1: Mode = 2, time_offset = 277.505493, sampleoff = 0.000137 freqoff = 0.366356
.....
initial -> 2.772697, tracking -> 1.887462, error -0.015296
initial -> 2.774176, tracking -> 1.888227, error 0.000799
initial -> 2.776244, tracking -> 1.888187, error -0.005090
initial -> 2.753825, tracking -> 1.888442, error -0.010391
initial -> 2.768445, tracking -> 1.888961, error 0.007719
initial -> 2.774968, tracking -> 1.888575, error 0.006177
initial -> 2.763738, tracking -> 1.888267, error -0.002878
initial -> 2.761683, tracking -> 1.888410, error -0.002655
initial -> 2.757870, tracking -> 1.888543, error -0.005434
initial -> 2.768815, tracking -> 1.888815, error 0.005062

```

Again, the differences are less than 2%.

The measurements were taken using the same recording of a transmission.

#### 4.5.2 Tracking the time offset

The effect of a timing error in sampling will be a phase shift over carriers with an increasing index within an ofdm-symbol. This phase shift is therefore an indicator for the time shift. Since successive carriers are uncorrelated, we need to fall back on successive pilot carriers within a symbol. We then look at the difference of the deviations of the phase at the pilot locations compared to the corresponding predefined pilot values. We compute for the most recent ofdm-symbol  $S$ ,

$$time_{offset} = \arg(\sum_{j=1}^{j=J} (carrier_{S,pilot_j} \times pilotValue_{S,pilot_j}^*) \times (carrier_{S,pilot_{j-1}} \times pilotValue_{S,pilot_{j-1}}^*)^*)$$

The basic assumption here is that the channel  $H_{n,pilot_j}$  is roughly equal to the  $H_{n,pilot_{j-1}}$  which might be the case in a slow fading environment when the distance - in number of carriers - between successive pilots is small (see e.g. [13] for a proof). For robustnessmode *B*, the distance is 6, the results seem to be good. Even for robustnessmode *A*, where between successive pilotcarriers in an ofdm-symbol is 20, the results are acceptable (note that robustnessmode *A* is meant for (almost) fading free transmissions).

While computing the tracking values, we also compute an estimate of the sampling clock offset, using [12] formula 5.40

$$\delta_{sc} = \frac{\sum_{j=0}^{J-1} \theta_j \alpha_j}{(2\pi \frac{T_s}{T_u})(\sum_{j=0}^{J-1} \alpha_j^*)}$$

where  $\alpha_i$  is the index of the  $i_{th}$  pilot.

**Implementation** The current time tracking value, is computed in the function *equalize*, member of the class *Equalizer\_1*. The offset is computed in a variable *offs2* and returned through a parameter.

The estimate of the sample clock offset is computed in the variable *offsa*, and returned through a parameter.

*Compensation* of the detected offset is in the function *getWord*, member of the class *ofdmProcessor*. Compensation is done by computing an offset value as the sum of the previous offset value with (a fraction of) the error, i.e.

$$timeOffset = timeOffset + 0.5 \times time_{offset}$$

*Correction* of the samples is by linear interpolation. Although it seems reasonable to expect that the  $\delta_{sc}$  can be used to reduce the timing error between successive samples, however, up until now no noticeable improvements showed.

**Results** Computing the time offset using the time domain symbols was tried by looking at the phase differences between the correlations of the cyclic prefix samples with the corresponding samples in the useful part. Results were disappointing. During the time that we are not (yet) synchronized, we use - as mentioned before - the time offset as computed by the *getMode* function, for each subsequent symbol adapted by the expected time shift caused by the sample clock offset.

There is a potential discontinuity when switching to the mainloop: the first time offset value maybe a little off, such that FAC (i.e. QAM4) demodulation can be done but the offsets are too much for reconstructing the QAM16 and QAM64 values. Computing an accurate time offset with time domain symbols is a topic for further study and experimentation.

The "quality" of the tracking of the time offset and the corresponding correction of the actual time offset can be deduced from the computed values for the time offset.

```
found 1: Mode = 3, time_offset = 91.789490, sampleoff = -0.000060 freqoff = -2.475263
time err = 0.000000, clock offset = 0.000000, correction value = 0.208478
time err = 0.001337, clock offset = -0.000015, correction value = 0.207810
time err = 0.000859, clock offset = -0.000033, correction value = 0.207380
time err = 0.002041, clock offset = -0.000045, correction value = 0.206360
time err = 0.001852, clock offset = -0.000053, correction value = 0.205434
.....
```

```

time err = 0.001944, clock offset = 0.000078, correction value = 0.175560
time err = 0.002364, clock offset = 0.000081, correction value = 0.174378
time err = 0.002139, clock offset = 0.000083, correction value = 0.173309
time err = 0.002788, clock offset = 0.000086, correction value = 0.171914
time err = 0.001835, clock offset = 0.000088, correction value = 0.170997
time err = 0.001459, clock offset = 0.000089, correction value = 0.170267

```

The table - derived from measurements on a recording - of radio Nigeria (Abidja) with robustnessmode C - shows the timing error, and the resulting timing offset. The time error remains reasonably small, the correction remains reasonably stable.

## 4.6 Equalization

The final step in the re-creation of the carrier values in the DRM frames, i.e. before decoding the bits, is the equalization, i.e. undoing all kinds of channel effects, such that the carriers in the frame resemble the carriers as they were sent.

A first - naive - approach of the equalizer was to compute the channels for the pilots with a simple algorithm, and apply - symbol per symbol - simple interpolation to create estimates of the channels for the other carriers and equalize them.

Algorithms for MLE and MSSE approaches to compute an "optimal" channel for the pilots exist. We experimented with one dimensional algorithms given in e.g. [16]. The results, combined with a variety of interpolation approaches, such as sync, fft etc - were somewhat disappointing, primarily due to issues around *extrapolation*. Extrapolation was needed to get estimates of the channels for the outermost carriers, outside the range of pilot carriers, in the symbols.

In diorama[14] an equalization algorithm is used that is based on a two dimensional approach[17]. Furthermore, the RXAMADRM software made by [15] contains a transliteration of the Diorama Matlab code in C++ code. That code was adopted as a basis for rewriting the equalizer module in the DRM decoder.

The basic idea for this kind of equalizer is to define for each carrier in a symbol a computation with which an estimate of the channel of that carrier is computed. The channel for that carrier is then computed by applying the computation to the pilot channels of the symbol the carrier is on and using the pilot channels on the symbols "not too far away" from the symbol the carrier is on.

Pilot channels themselves are easily estimated by comparing the received value with the value they should have. In our implementation we default to the simple one, i.e. a quotient of the observed value and the predefined value<sup>7</sup>.

Shortwave - where the DRM programs are transmitted - suffers from severe fading conditions. Figure 1 shows - with the red lines in the box in the bottom half - the amplitude correction computed by the equalizer for a sequence of frames. Each line (two lines from each frame are depicted) in that picture indicates the amplitude correction to be applied to the carriers in a symbol of a frame.

We compute the channel for a given carrier *carrier* in a given ofdm-symbol *symbol* as

$$channel_{symbol,carrier} = \sum_{p \in pilots} channel_{pilot(p)} \times W\_symbol\_blk_{symbol,carrier}(p)$$


---

<sup>7</sup>In the implementation there is the possibility - a configuration option - for computing the channels of the pilots using MLE techniques as mentioned in [16]

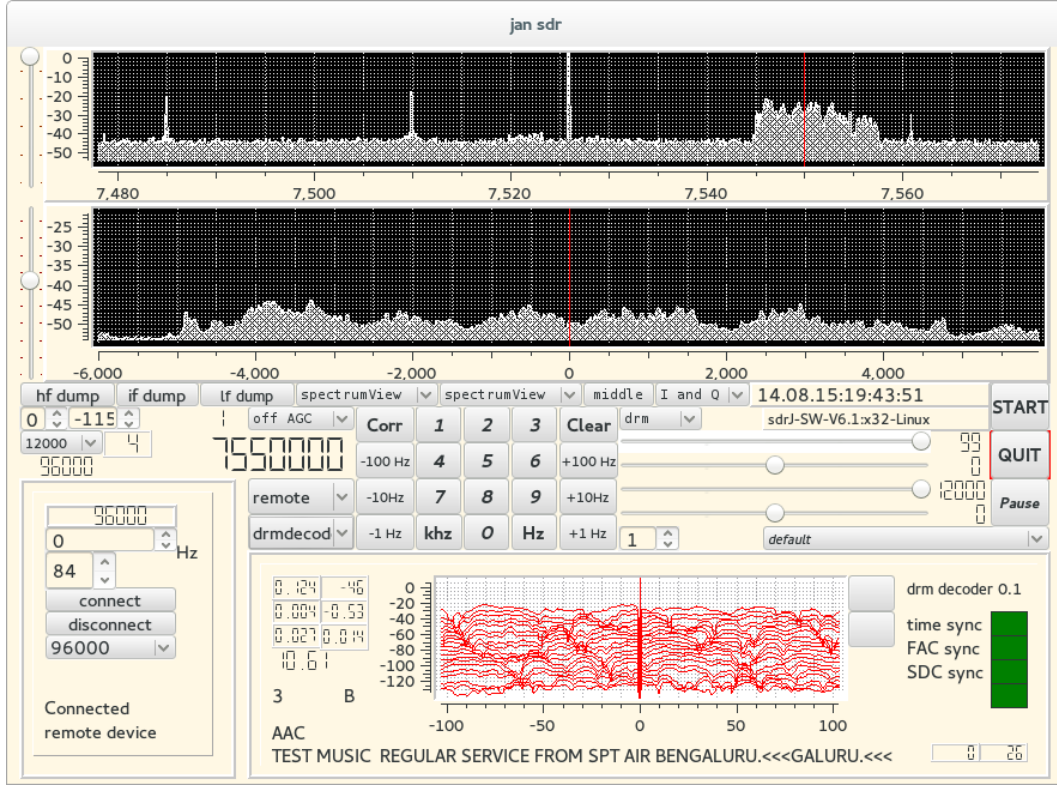


Figure 1: Amplitude correction by equalizer (2 "spaghetti" lines per frame)

i.e. a convolution of pilot values "nearby" the carrier under consideration with a set of pre-computed values. In the above formula,  $pilots$  indicates the set of pilots of all ofdm-symbols taken into account (i.e. the pilots "nearby"), and  $channel_{pilot(p)}$  denotes the channel value for the  $p$ -th  $pilot(p)$  in this set.

$W_{symbol\_blk_{symbol,carrier}(p)}$  denotes the  $p$ -th element of the predefined values for convolving with the  $p$ -th real pilot value used for computing the channel for carrier  $carrier$  in symbol  $symbol$ .

*Equalization*, i.e. restoring the carrier value, is then straightforward,  

$$\frac{actualValue_{symbol,channel}}{channel_{symbol,carrier}}.$$

**Computing  $W_{symbol\_blk}$**  It is important to realize that the pilot pattern repeats itself, after *periodicity* symbols pilots are found on the same carrier positions<sup>8</sup>, however, the pilot values on corresponding positions in different symbols may differ.

The value of *periodicity* depends on the robustnessmode, for e.g. mode A, the periodicity is 5, for mode B it is 3, for mode C it is 2.

In the approach taken, the computation of the channel for a carrier  $c$  in ofdm-symbol  $s$

<sup>8</sup>Not taking into account the additional pilots for "special purposes" in symbol 0.

depends on the pilot values in the symbols  $s - a + 1..s + a$  (where  $a$  is essentially user defined). We use the term *Window* for such a range of  $2 \times a$  symbols. The symbol  $s$  is then the base symbol of that Window.

It is obvious that there are only *periodicity* different windows in frames of a given mode (not taking into account different pilot values on corresponding positions in different symbols within a frame).

Given that there are *periodicity* different windows in a frame, it will be obvious that - apart from the particular parameter values - the computation of the channels in  $symbol_i$  is the same as the computation of the channels in  $symbol_{i+k*periodicity}$ , for  $i + k * periodicity < numberOfSymbols(mode)$ .

$W\_symbol\_blk$  will be implemented as a matrix, with dimensions ( $Windows \times carriers \times pilots\_per\_window$ ).

- The first dimension is the number of Windows, i.e. *periodicity*, indices have a value in the range  $0..periodicity$ ,
- The second index indicated the order number of the carrier in the base symbol of the window,
- the third index gives access to a vector with coefficients for convolving with the real pilot channels. The number of pilots in the different windows may differ.

With such a matrix, computation of the channel for a carrier  $c$  in a symbol  $s$  is simple:

$$\sum_{i=0}^{NumberOfPilots(W(s))} W\_symbol\_blk[W(s)][c][i] * pilotValue_{realAddress[s][Table[W(s)][i]]}$$

In this formula:

- $W(s)$  is the mapping from the actual symbol to a window;
- $NumberOfPilots(x)$  gives the number of pilots in Window  $x$ ;
- $Table[x][y]$  gives the address of the  $y$ -th pilot in window  $x$ , i.e an address relative to the first carrier of the first symbol of that window;
- $realAddress[s][y]$  gives the real address, i.e. *symbol*, *carrier*, of the  $y$ -th pilot in window  $x$ , applied on symbol  $s$ .

Following [17], computing the coefficients requires computing the cross covariance matrix  $\Theta$  and the autocovariance  $\Phi$ . Given the  $\Theta$  and the  $\Phi$ , we compute for each window  $W$  and each carrier  $c$  in the base symbol of that window

$$W\_symbol\_blk_{w,c,pilot} = \sum_{k \in pilots_w} \Theta_{w,c,k} \times \Phi_{k,pilot}^{-1}$$

with

$$\Theta_{w,c,pilot} = sinc((c - carrier_{pilot}) \times f_{cut_k}) \times sinc((periodicity/2 - symbol_{pilot}) \times f_{cut_t})$$

where  $pilots_{window}$  denotes the set of pilot locations (each pilot addressed as (symbol, carrier)),  $carrier_{pilot}$  indicates the (*symbol*, *carrier*) position of the pilot, *periodicity* indicates the afore mentioned periodicity of the pilot patterns in the symbol, *symbol* is the (relative) number of the symbol carrying the pilot.

$f_{cut_t}$  is an estimate for the two-sided maximum doppler frequency (normalized w.r.t. symbol duration  $T_s$ ), and

$f_{cut_k}$  is an estimate for the two-sided maximum echo delay (normalized w.r.t useful symbol duration  $T_u$ ).

$\Phi$  is computed over all regular pilots appearing in a window, and is - per window - a 2-dimensional array, with:

$$\Phi(pilot_1, pilot_2) = \text{sinc}((carrier_{pilot_1} - carrier_{pilot_2}) \times f_{cut_k}) \times \text{sinc}((symbol_{pilot_1} - symbol_{pilot_2}) \times f_{cut_t})$$

$carrier_{pilot_i}$  indicates the carrierposition of  $pilot_i$ .  $symbol_{pilot_i}$  indicates the (relative) ofdm-symbol number where the pilot resides (relative to the first symbol of the window we are dealing with).

**Implementation** The implementation in a class *Equalizer\_1* follows the outline given above. The constructor of the class will precompute  $W\_symbol\_blk$  along the lines, mentioned above.

The quality of the equalization depends (a.o) on the size of the windows. Since the amount of pilots used in the computation depends on the window size, larger windows mean more computations per element. To experiment, one of a couple of different sizes for the windows (i.e. the number of symbols for which the pilots are used in the window) can be selected. Selection is through a setting in the ".ini" file.

Values  $f_{cut_k} = 0.0675 \times \frac{1}{windows}$  and  $f_{cut_t} = 1.75 \times \frac{T_g}{T_u}$  are taken from [14] and adapted for minimal disturbance of the equalizers (i.e. different procedure for different robustnessmodes).

The actual estimators used for estimating the channel values of the pilots are selected by a configuration parameter, the trivial one does not need additional matrix operations, the MLE based ones require the armadillo library[18] for the matrix operations involved. By default the trivial one is selected<sup>9</sup>

$$channel_{pilot_j} = \frac{actualValue_{pilot_j}}{definedValue_{pilot_j}}$$

The estimators are implemented in instances of a separate *Estimator\_1* class.

For each window  $W$  the - relative - locations of the pilots in the window are computed by the function *buildTrainers*. The locations of the trainers (addressed as pair (symbol, carrier)) are stored in a vector, addressable through *theTrainers[window]*. Once these "trainers" are built, for each window the  $\Phi$  matrix is computed as indicated above in nested loops.

In the same loop for each carrier the  $\Theta$  is computed - again the implementation follows the description above - and for that carrier  $c$  the vector  $W\_symbol\_blk[W][c]$  is computed.

The actual equalization for a carrier then takes as steps:

- identifying the vector in  $W\_symbol\_blk$  that is applicable for the carrier;
- convolving the actual pilot values of the window under consideration with the the value (a vector) of  $W\_symbol\_blk$  to obtain the channel for the carrier under consideration.
- applying the channel value to equalize the carrier.

Note that the channel values are stored; they are used in the symbol decoding later on.

The vector to be equalized is placed in the *Frame*, computations for tracking values are done, and - within the function *processSymbol*, the actual computation is done (using the channel values on the pilot locations).

The result is stored in a row in the two-dimensional array *outFrame*.

---

<sup>9</sup>It is obvious that lots of experiments were done with the other approaches as well, however, differences in performance were not noticeable

## 5 Final remarks

In order to get a better understanding of the structure of DRM, both the Matlab code of Diorama[14] and the C++ transliteration of Diorama in C++ as done in RXAMADRM [15] were looked at. Both proved to be a great help. Some parts of the DRM decoder are a rewrite of the C++ code in RXAMADRM, other parts were - at least - inspired by Diorama and RXAMDRM.

The SDR-J software uses a number of libraries, made available through (L)GPL style licenses and parts of the code is based on ideas of others. In all cases attempts are made to indicate the rightful owner of the copyrights. The software itself is available as is, under a GPL V2 license.

The SDR-J software is a hobby project, its development is neither driven by time constraints nor deadlines, but purely by curiosity. It allows to re-implementations of sections, something that has been done regularly. The software is - obviously - not finished, after all it is software and as far as I am concerned, it never will be finished. Many enhancements (and experiments) are still waiting to be done. Support in any form - suggestions, code improvements or donations - is always welcome.

## References

- [1] Jan van Katwijk, SDR-J-SW-7.1, Software for SDR : The 32 bits SW receiver, see "<http://www.sdr-j.tk>"
- [2] Burkhard Kainka, Software Defined Radio Met USB-interface, Elektor 5/2007.
- [3] Alberto, I2PHD, Winrad - specifications for the external I/O DLL, Document found on the internet.
- [4] see "<http://www.sdrplay.com>"
- [5] see "<http://www.iw3aut.altervista.org/>"
- [6] see "[www.fftw.org](http://www.fftw.org)"
- [7] see "[www.audiocoding.com/faad2.html](http://www.audiocoding.com/faad2.html)" and "[http://drm.sourceforge.net/wiki/index.php/Instructions\\_for\\_building\\_a\\_version\\_suitable\\_to\\_handle\\_DRM\\_aac\\_samples](http://drm.sourceforge.net/wiki/index.php/Instructions_for_building_a_version_suitable_to_handle_DRM_aac_samples)"
- [8] see "[www.portaudio.com](http://www.portaudio.com)"
- [9] see "<http://www.mega-nerd.com/libsndfile/>"
- [10] Mérouane Debbah, Short introduction to OFDM, Document found on the internet.
- [11] Final draft ETSI ES 201 980 V3.1.1 (2009-06) ETSI Standard Digital Radio Mondiale (DRM); System Specification.
- [12] Tzi-Dar Chiueh, Pei-Yun Tsai, OFDM Baseband Receiver Design for Wireless Communications, John Wiley and Sons (Asia) Pte Ltd, 2007.



- [13] Dah-Chung Chang, Effect and Compensation of Symbol Timing Offset in OFDM systems with Channel Interpolation, IEEE Transactions on Broadcasting, Vol 54, 4, December 2008.
- [14] T. Schorr, A. Dittrich, see "<http://nt.uni-kl.de/forschung/diorama>".
- [15] Ties Bos - PA0MBO, RXAMADRM - A Linux program for digital SSTV, November 2014. Found on the internet.
- [16] Michele Morelli, Umberto Mengali, A Comparison of Pilot Aided Channel Estimation Methods for OFDM Systems, IEEE Transactions on Signal processing, Vol 49 (12), December 2001.
- [17] Peter Hoeher, Stefan Kaiser, Patrick Robertson, Two dimensional Pilot-Symbol-Aided Channel Estimation By Wiener Filtering, in Proc. IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP), Munich, Germany, Apr. 1997, pp. 1845-1848.
- [18] see: "<http://arma.sourceforge.net/>"